Amendments to the Claims

This listing of claims will replace all prior versions, and listings, of the claims in the application. All claims currently being amended are shown with deleted text struckthrough or double bracketed and new text underlined. Additionally, the status of each claims is indicated in parenthetical expression following the claim number.

Claims 1 – 23, and 25 – 30 remain.

Claims 2, 16, 22, and 23 are being amended.

Claim 24 is being cancelled.

WHAT IS CLAIMED IS:

1. (Original) Digital tone controls comprising:

a first path including a digital filter and a scaler for controlling a level of a low frequency component of a received digital audio signal;

a second path including a digital filter and a scaler for controlling a level of a high frequency component of the received digital audio signal;

a third path including a scaler for controlling a level of an unfiltered component of the received audio signal; and

a summer for adding a contribution from each of the paths to generate a composite signal having a selected gain -- frequency response.

2. (Currently Amended) The digital tone controls of Claim 1 wherein the digital filters comprise [[a]] infinite impulse response filters.

3. (Original) The digital tone controls of Claim 1 wherein the digital filters comprise finite impulse response filters.

4. (Original) The digital tone controls of Claim 1 wherein the filters and scalers are implemented in software.

5. (Original) The digital tone controls of Claim 1 wherein the filters and scalers are implemented in software executed on a digital signal processor.

6. (Original) The digital tone controls of Claim 1 wherein the digital filters are first order filters.

7. (Original) The digital tone controls of Claim 1 wherein the scalers multiply the filter output by a positive coefficient.

8. (Original) A method of controlling tonal level in a digital audio data stream comprising the steps of:

filtering the audio data stream with a plurality of digital filler to extract a plurality of frequency components of a selected set of frequency bands;

selectively scaling each of the frequency components;

scaling an unfiltered component of the digital audio data stream; and

summing the scaled frequency components with the scaled digital audio data stream to generate a digital signal having a selected frequency response.

9. (Original) The method of Claim 8 wherein said step of filtering comprises the substeps of:

filtering the audio data stream with a bass low pass filter passing frequencies below a first corner frequency;

filtering the audio data stream with a treble low pass filter passing frequencies below a second corner frequency, the second corner frequency being higher in frequency than the first corner frequency;

filtering the audio data stream with a bass high pass filter passing frequencies above a third corner frequency; and

filtering the audio data stream with a treble high pass filter passing frequencies above a fourth corner frequency, the fourth corner frequency being higher in frequency than the third corner frequency.

10. (Original) The method of Claim 8 wherein said step of filtering comprises the step of filtering the audio data stream using software filters.

11. (Original) The method of Claim 8 wherein said step of filtering comprises the step of filtering the audio data stream with a first order digital filter.

12. (Original) The method of Claim 8 wherein said step of filtering comprises the step of extracting each frequency component with an infinite impulse response filter.

13. (Original) The method of Claim 8 wherein said step of filtering comprises the step of extracting each frequency component with a finite impulse response filter.

14. (Original) The method of Claim 13 wherein the finite impulse response filter is of a 2^{nd} order or greater.

15. (Original) The method of Claim 13 wherein the scalers take on positive values only.

16. (Currently Amended) An audio processing device comprising:

a port for receiving a stream of audio data; and

a digital signal processor operable to:

filter out and scale a low frequency component of a data stream extracted from said received stream of audio data;

filter out and scale a high frequency component of the extracted data stream;

scale an unfiltered component of the extracted data stream; and

add[[ing]] the scaled low and high frequency components and the scaled unfiltered component of the extracted data stream to generate a composite signal having a selected gain – frequency response.

17. (Original) The audio processing device of Claim 16 wherein said digital signal processor comprises a selected one of a plurality of digital signal processors forming an audio decoder.

18. (Original) The audio processing device of Claim 16 wherein said digital signal processor is operable to execute program code implementing infinite impulse filters for filtering out said low and high frequency components of said extracted data stream.

19. (Original) The audio processing device of Claim 16 wherein said digital signal processor is operable to execute program code implementing finite impulse response filters for filtering out said low and high frequency components of said extracted data stream.

20. (Original) The audio processing device of Claim 16 wherein said digital signal processor is operable to execute program code implementing multipliers for scaling said components of said extracted data stream.

21. (Original) The audio processing device of Claim 16 wherein said received audio data stream comprises a compressed audio data stream and said extracted data stream extracted from said received data stream comprises a stream of PCM data.

22. (Currently Amended) The audio processing device of Claim 16 wherein said received audio data stream comprises an uncompressed data stream and said extracted date stream comprises a stream of PCM data.

23. (Currently Amended) A program for implementing tone controls in a programmable audio processing device comprising:

a plurality of digital filters having programmable coefficients for extracting a plurality of frequency components from a digital audio data stream;

a plurality of programmable scalers each for selectively setting an amplitude of one of the frequency components output from a corresponding digital filter; [[and]]

a programmable scaler for selectively setting an amplitude of an unfiltered component of the digital data stream to generate a scaled unfiltered component;

a summer for summing a plurality of scaled frequency components output from the programmable scalers and the scaled unfiltered component

24. Cancelled

25. (Original) The program of Claim 23 wherein the coefficients are selected to extract at least one bass component and at least one treble component from the audio data stream.

26. (Original) The program of Claim 23 wherein the coefficients are selected to provide a plurality of filters including:

a bass low pass filter extracting frequencies below a selected bass frequency;

a bass high pass filter for extracting frequencies above a selected bass frequency;

a treble low pass filter for extracting frequencies below a selected treble frequency; and

a treble high pass filter for extracting frequencies above a selected treble frequency.

27. (Original) The program of Claim 23 wherein the digital filters are first order.

28. (Original) The program of Claim 23 wherein the digital filters are second order or higher.

29. (Original) The program of Claim 23 wherein the digital filters are IIR filters.

30. (Original) The program of Claim 23 wherein the digital filters are FIR filters.